스피커의 저주파 대역 왜곡 측정의 새로운 기법

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A New Method for Measuring the Distortions of the Loudspeaker at Low Frequencies

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Abstract

A method to measure the distortions of the loudspeaker at low frequencies without an anechoic chamber is proposed. This method utilizes the fact that the n-th harmonic distortion outside the enclosure is boosted by 40log$n$ dB compared to that inside the enclosure. By compensating for the effect of standing waves occuring inside the enclosure, it is possible to predict the distortions for wide frequency ranges below the first break-up frequency of the diaphragm. The method is applicable to the intermodulation distortion as well.

1. Introduction

In assessing the loudspeaker quality objectively, it is most essential to measure the frequency response and distortions. To measure the frequency response and the distortions, it is customary to use an anechoic chamber since it is necessary to pick up the sound pressure of the loudspeaker itself without reverberation. However, measuring the sound pressure in the anechoic chamber is expensive. In addition, the anechoic chamber has its lower cutoff frequency below which the measurement data lose reliability.

R. H. Small[1] has devised a method to predict the frequency response and the total harmonic distortion outside the enclosure from the measurement of internal pressure. The method was applied in measuring the volume velocity of a loudspeaker[2] and in testing home loudspeakers[3]. However the effect of standing waves inside the enclosure was not manipulated in any way but there were tries to find an optimum measuring position where the standing waves are minimum, so that the frequency range applicable was very low such as below 200Hz.

In this paper, the relation between the distortion level inside the enclosure and the distortion level outside the enclosure is first described. The standing wave measured for the frequency response is used as a clue in predicting the distortion levels. Then the effect of standing waves inside the enclosure is compensated for. This makes it possible to predict the harmonic and intermodulation distortion up to reasonably high frequencies without an anechoic chamber.

2. Relation of the distortions between inside and outside the enclosure

For adiabatic process, the total volume and the pressure within an enclosure obey the relation[4]

$$(p_0 + p_{in})(V_0 + \Delta V)^{\gamma} = p_0 V_0^{\gamma} \ , \tag{1}$$

where γ is the ratio of the specific heat of the gas at constant pressure to the specific heat at constant volume for the gas. V_0 and p_0 are the undisturbed volume and static pressure respectively, and p_{in} and ΔV are the incremental pressure inside the enclosure and incremental volume, respectively.

Applying the relation $\Delta V = Sx$, where S is the effective diaphragm area of a loudspeaker and x is the displacement of the diaphragm, and rewriting eq.(1) with respect to x, eq.(2) relating the pressure p_{in} inside the enclosure and the displacemant of the diaphragm x is obtained.

$$x = \frac{V_0}{S}\left[\left(1 + \frac{p_{in}}{p_0}\right)^{-\frac{1}{\gamma}} - 1\right]$$

$$\cong -\frac{V_0}{S\gamma p_0}\left(p_{in} - \frac{\gamma+1}{2\gamma p_0}p_{in}^2\right) . \tag{2}$$

The 2nd term of the right side of eq.(2) is the distortion of the medium itself. This distortion can be easily calculated if we measure the SPL inside the enclosure. That is, the level of air distortion is (2×SPL-195) dB re 20 μPa. For example, if SPL inside is 140dB the level of air distortion is 85 dB. As long as this air distortion is far smaller than the distortion of a system, we can neglect the air distortion and consider the air a linear medium[5]. Then x may be considered to be proportional to p_{in}, i.e.,

$$x \cong -\frac{V_0}{S\gamma p_0}p_{in} \ . \tag{3}$$

For a baffled piston radiator vibrating with radian frequency ω and displacement x, far field sound pressure $p_{out}(r,\theta)$ is[6]

$$p_{out}(r,\theta)=-\frac{\rho_0 S}{2\pi r}\omega^2 x\left[\frac{2J_1(ka\sin\theta)}{ka\sin\theta}\right]e^{j(\omega t-kr)}\ ,\qquad(4)$$

where r and θ are the distance and the angle to the measuring position respectively, k wave number, a the radius of the piston, and $J_1(\cdot)$ the Bessel function of the first kind of order 1. The bracketted term determines the directional characteristics of the piston radiator.

If x is not pure sinusoidal but distorted to produce the harmonics, x can be denoted as

$$\begin{aligned}x&=\sum_{n=1}x_n\\&=\sum_{n=1}\hat{x}_n\cos(n\omega t+\phi_n)\ ,\end{aligned}\qquad(5)$$

where n is a harmonic number, ϕ_n phase of the n-th harmonic x_n, $\hat{x}_n$ amplitude of x_n, respectively. In this case, sound pressure outside the enclosure is found to be

$$\begin{aligned}p_{out}(r,\theta)&=-\frac{\rho_0 S}{2\pi r}\sum_{n=1}(n\omega)^2 x_n\left[\frac{2J_1(nka\sin\theta)}{nka\sin\theta}\right]e^{j(n\omega t-nkr+\phi_n)}\\&=-\frac{\rho_0 S\omega^2}{2\pi r}\sum_{n=1}n^2 x_n\left[\frac{2J_1(nka\sin\theta)}{nka\sin\theta}\right]e^{j(n\omega t-nkr+\phi_n)}\end{aligned}\qquad(6)$$

If the sound pressure is measured on-axis, eq.(6) is reduced to

$$p_{out}(r,0)=-\frac{\rho_0 S\omega^2}{2\pi r}\sum_{n=1}n^2 x_n e^{j(n\omega t-nkr+\phi_n)}\ .\qquad(7)$$

Applying eq.(3) to eq.(7) for each harmonic, n-th harmonic outside the enclosure, $p_{out,n}$, is related with n-th harmonic inside the enclosure $p_{in,n}$ as

$$\left|p_{out,n}\right|=\frac{\rho_0 V_0\omega^2}{2\pi\gamma p_0 r}n^2\left|p_{in,n}\right|\ .\qquad(8)$$

The sound pressure level outside is thus related to the sound pressure level inside the enclosure as follows:

$$SPL_{out,n}=SPL_{in,n}+40\log\omega+40\log n+C\ ,\qquad(9)$$

where $C=20\log(\rho_0 V_0/2\pi\gamma p_0 r)$ is the constant determined by the inner volume V_0 of the enclosure and the distance r to the measuring position and all of the physical quantities are in SI units. Eq.(9) reveals that the n-th distortion outside the enclosure is 'boosted' by 40log$n$ dB compared to that inside. For example, the 2nd distortion is boosted by 12 dB and the 3rd distortion is boosted by 18 dB outside the enclosure, respectively.

Eq.(4) is also used in predicting the intermodulation distortion outside the enclosure. That is, if ω_{IM} is the intermodulation angular frequency of interest, the on-axis pressure relation of the intermodulation distortion between inside and outside the enclosure is as follows:

$$p_{out,IM}=-\frac{\rho_0 S}{2\pi r}\omega_{IM}^2 x_{IM}e^{j(\omega_{IM}t-k_{IM}r)}\ ,\qquad(10)$$

where x_{IM} is the intermodulated component of displacement vibrating with angular frequency ω_{IM}. Applying eq.(3) to eq.(10) and denoting the relation of the pressure of intermodulation distortion between inside and outside the enclosure as sound pressure level, the relation

$$SPL_{out,IM}=SPL_{in,IM}+40\log\omega_{IM}+C\qquad(11)$$

is obtained.

3. Standing wave inside the enclosure

If the vibrating frequency of the diaphragm of the loudspeaker is so high that the wavelength is comparable or shorter than the dimension of the enclosure, standing wave builds up inside the enclosure. The spatial pressure distribution is determined by 1) the dimension and the shape of the enclosure, 2) treatment of the damping material, 3) vibrating frequency of the diaphragm, and 4) the source distribution. For a specific loudspeaker enlosure, 1) and 2) are fixed and only 3) and 4) determines the spatial pressure distribution.

As long as the diaphragm vibrates as a rigid body, i.e., it does not break up, the spatial pressure distribution due to the fundamental of frequency f and that due to the n-th harmonic of frequency f, whose fundamental is f/n, will be identical (Fig.1(a)). The reason is that the frequency and the source distributions of the two cases are identical. In that case, we can make use of the standing wave as a clue to compensate for the fluctuation occuring at high frequencies. The procedure is as follows:

1) First, we measure the frequency response H(f) inside the enclosure. H(f) contains the effect of standing waves. We let this effect of standing wave be the transfer function R(f). Fig.4 is an example of R(f).
2) The theoretical frequency response $H_{th}(f)$ inside the enclosure is calculated. $H_{th}(f)$ can be simply calculated by measuring the resonance frequency f_0 and the Q-factor of the loudspeaker.
3) The difference between H(f) and $H_{th}(f)$ is R(f). That is, R(f)=H(f)-$H_{th}(f)$ in logarithmic scale.
4) We measure the n-th harmonic distortion $D_n(f)$. As in the case of H(f), $D_n(f)$ contains the effect of standing wave R(f).
5) Since it is necessary to exclude the effect of standing wave to predict the distortions, R(f) should be excluded from $D_n(f)$. Therefore, $D_n(f)$-R(f) results in the correct prediction of n-th harmonic distortion (In practice, the graph of the distortion is plotted along the fundamental frequency scale, e.g., $D_n(f)$ is plotted as $D_n(f/n)$. Therefore, R(f) should also be rescaled as R(f/n) and $D_n(f/n)$-R(f/n) is the prediction of the distortion.).

This process makes it possible to predict the distortions of the loudspeaker.

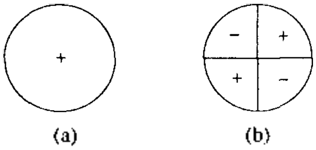

Fig.1. Examples of the source distributions producing the same frequency f. (a) the diaphragm vibrating with fundamental f/n, which is below the first break-up frequency (The n-th harmonic frequency of it corresponds to f). (b) the diaphragm vibrating at fundamental f at which it breaks up.

If, however, the diaphragm does not move in-phase, that is, the diaphragm breaks up as shown in Fig.1 (b), or if the amplitude of vibration is not the same everywhere at the diaphragm, which occurs near the first break-up frequency, the relations of eqs.(9) and (11) do not hold and the prediction will be incorrect. The prediction is thus valid only at low frequency range below the first break-up frequency where it is possible to assume that the diaphragm moves as a rigid piston.

4. Experimental result

Sound pressure inside the enclosure is measured with the set-up shown in Fig.2. The dimension of the loudspeaker under measurement is 16.9×21.1×35.7cm and the diaphragm diameter is 10cm. The thickness of the enclosure panel is 9mm, which is somewhat thin. A condenser microphone B&K 4133 is placed inside the closed-box loudspeaker enclosure and special care is taken to make the enclosure air-tight. A $4V_{rms}$ sweep signal is applied to the loudspeaker from 20 Hz to 1kHz, which is thought to be low enough compared to the first break-up resonance frequency. The first break-up frequency is calculated as high as 12kHz with the loudspeaker under test[7].

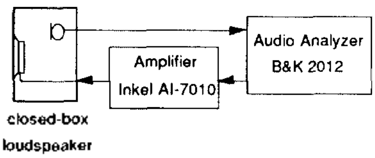

Fig.2. Set-up for measuring the sound pressure level inside the loudspeaker enclosure.

Fig.3 shows the fundamental, the 2nd harmonic, the 3rd hamonic, which have been measured inside the enclosure, and the air distortion level, respectively. At low frequency ranges below about 300 Hz, the fundamental in Fig.3 can be considered to represent the curve of the displacement of the diaphragm since eq.(3) holds and the fundamental is dominant in total pressure. One thing to note is that there exist fluctuations over 300Hz of the fundamental. If there would be no standing wave inside the enclosure, the slope of the fundamental above the loudspeaker resonance frequency would be -40 dB/decade[8]. The fluctuations are also appeared in the 2nd and 3rd harmonic. Furthermore, we can see the similarity between the fluctuation of the fundamental and those of the distortions. The difference between the fundamental in Fig.3 and the theoretically calculated one is the transfer function $R(f)$ resulting from the standing wave inside the enclosure(Fig.4).

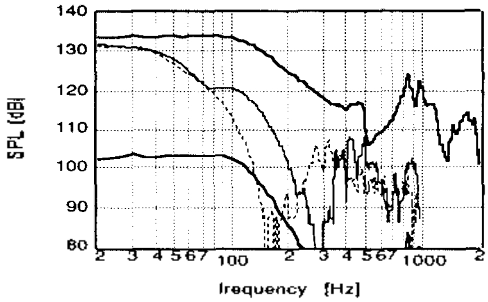

Fig.3. Sound pressure levels measured inside the loudspeaker. Upper thick solid line : fundamental. Thin solid line : 2nd harmonic. Dashed line : 3rd harmonic. Lower thick solid line : air distortion level. The 2nd and 3rd harmonic and air distortion level are raised up by 30dB for graphical presentation.

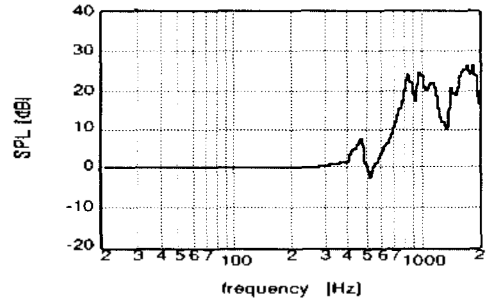

Fig.4 Transfer function R(f). This is due to the standing waves inside the loudspeaker enclosure.

Using the relation of eq.(9), Fig.4 is converted to Fig.5. In order to obtain the correct prediction of the distortions, the effect of standing wave should be compensated for. To do this, the frequency axis of $R(f)$ is rescaled as $R(f/2)$ for the 2nd distortion and as $R(f/3)$ for the 3rd distortion and then they are subtracted from the 2nd and 3rd harmonic distortions shown in Fig.5 respectively. This process corresponds to the compensation of the effect of standing wave.

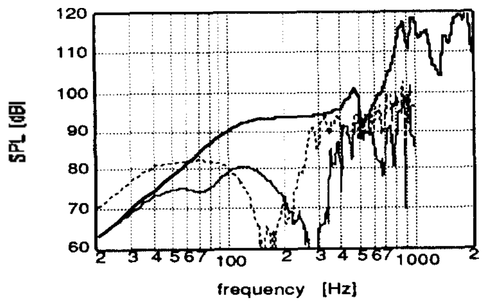

Fig.5. Frequency response and the 2nd and 3rd distortion predicted by eq.(9). Thick solid line : fundamental. Thin solid line : 2nd harmonic. Dashed line : 3rd harmonic(2nd and 3rd harmonic raised up by 20dB).

After compensating for the 2nd and 3rd harmonic distortions, we get Fig.6. Since the effect of standing wave is removed, Fig.6 is the prediction of the 2nd and 3rd distortions.

Frequency response and distortions measured in an anechoic chamber are shown in Fig.7 for comparison. Comparing Fig.6 and Fig.7, we can see that the 2nd and 3rd distortion's prediction shows reasonably good agreement with those measured in the anechoic chamber. In the case of the 2nd distortion, the prediction and the measured data agrees well within a ±3dB deviation up to 1kHz except between 270Hz and 350Hz. As for the prediction of the 3rd distortion, however, there is some large discrepancy above 150Hz.

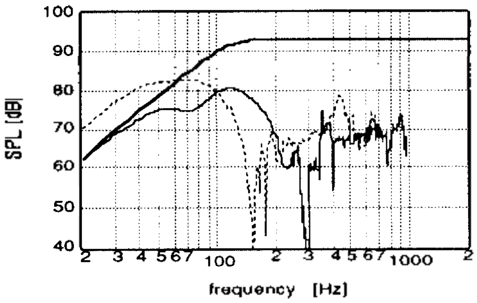

Fig.6. 2nd and 3rd distortion predicted by applying eq.(9) and compensating for the effect of standing wave inside the loudspeaker. Thick solid line : fundamental. Thin solid line : 2nd harmonic. Dashed line : 3rd harmonic(2nd and 3rd harmonic raised up by 20dB).

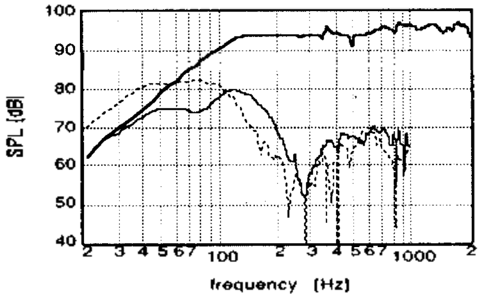

Fig.7 .Frequency response and the 2nd and 3rd distortion measured in an anechoic chamber.

The causes of discrepancy are mainly the air distortion[5] and the enclosure vibration[9]. The amount of air distortion depends on the pressure level. If the level of the air distortion exceeds the level of the diaphragm distortion, the diaphragm distortion cannot be measured correctly. The level of air distortion is (2×SPL-195) dB and if this exceeds the 2nd and 3rd harmonic distortion in Fig.3, a correct prediction is impossible. The lower thick solid line in Fig.3 denotes the level of air distortion. In the case of the 2nd distortion, the diaphragm distortion level is below the air distortion level in 270Hz to 300Hz and in the case of the 3rd distortion, 130Hz to 190Hz. Therefore, In these frequency ranges respectively, the prediction has large discrepancy.

In order to analyze the effect of enclosure vibration, the accelerations at the front panel where the driver is attached and at the side panel are measured respectively. The fundamental of the accelerations is shown in Fig.8. As expected, we can see that the vibration of the enclosure corrupts the frequency response. The effect of the panel vibration applies to the distortions as well. In Fig.9, the 2nd distortion of panel acceleration is overlaid on the 2nd distortion predicted. We can see that the corruption of the 2nd distortion is more severe than that of the fundamental. And it is found that the 2nd distortion of the panel is high between 250Hz and 400Hz, which proves the deviation in that range is mainly due to panel distortion.

Intermodulation distortions are likewise predicted from the measurement of the internal pressure.

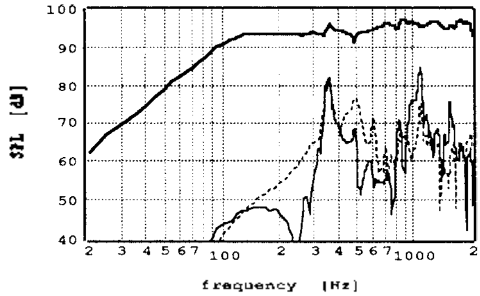

Fig.8. Frequency response affected by the vibration of the enclosure. Thick solid line : fundamental. Thin solid line : acceleration of the front panel where the driver unit is attached. Dashed line : acceleration at the side panel of the enclosure. Accelerations are in arbitrary reference.

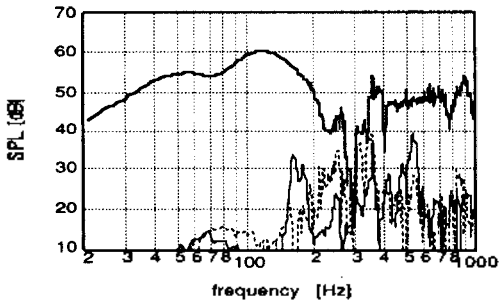

Fig.9. 2nd distortion affected by the vibration of the enclosure. Thick solid line : 2nd distortion predicted. Thin solid line : 2nd distortion of acceleration at the front panel. Dashed line : 2nd distortion of acceleration at the side panel of the enclosure.

5. Conclusion

A method to predict the distortions of the closed-box loudspeaker without an anechoic chamber is proposed. This method utilizes the fact that the n-th harmonic distortion is boosted by $40\log n$ dB outside the enclosure compared to that of inside the enclosure. Since the standing waves take place not only by the fundamental but also by the distortions in high frequency ranges, the effect of standing waves can be compensated for.

The experiment shows fairly good agreement between the predicted curve and the measured curve in the anechoic chamber. The proposed method applies not only to the harmonic distortions but also to the intermodulation distortion.

The advantages of this method are:

1) It is not necessary to measure in an anechoic chamber. Only a small space is sufficient.
2) Outer noise does not matter.
3) It is simple to apply since it is not necessary to find an optimum position of measurement inside the enclosure.
4) Damping treatment inside the enclosure causes no problem since the internal resonance phenomenon is a function of the frequency that the microphone accepts and therefore standing wave compensation is possible in any condition.

On the other hand, there are some limitations in applying this method:

1) The prediction is valid only for the closed-box type loudspeaker and low frequencies below the first break-up frequency.
2) The enclosure should be hard enough since the enclosure vibration gives error to the prediction.
3) The measuring microphone should be of high pressure type since the pressure inside the enclosure is fairly high.

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